BIOGRAPHICAL INFORMATION

Raymond L. Pickholtz	Professor and Past Chairman Department of Electrical Engineering and Computer Science The George Washington University Telephone: (202) 994-6538 FAX (202) 994-0227
	President: Telecommunication Associates Telephone: (703) 280-5628
Fields of Specialization:	Communication Systems Data and Computer Communications Telecommunication Networks Secure Communications Array Signal Processing
Professional Background:	
1964 - 1971	Adjunct Associate Professor Physics, Brooklyn College of CUNY
1972 - Present	Professor of Engineering & Applied Science, The George Washington University
1977	Visiting Professor, Universite du Quebec, Institut National Researches de la Scientifique
1978 - 1980	Chairman, Department of Electrical Engineering and Computer Science, The George Washington University
1983, 1990	Visiting Professor, University of California at San Diego, Depart. of Elect. Eng. & Computer Sci. at Santa Barbara, Depart. of Computer Sci.
1965 - Present	President, Telecommunications Associates, Inc.

1954 - 1960	New York University, Instructor of Engineering Electronics
1954 - 1957	RCA Laboratories (Research on Color Television Circuitry and Systems)
1957 - 1959	ITT Laboratories - Senior Engineer
1959 - 1962	ITT Laboratories -Technical Specialist
1962 - 1965	Polytechnic Institute of Brooklyn, Instructor in Electrical Engineering
1965 - 1967	Assistant Professor of Electrical Engineering, Polytechnic Institute of Brooklyn
1967 - 1971	Associate Professor of Electrical Engineering, Polytechnic Institute of Brooklyn
1968 - 1970	Consultant, IBM T.J. Watson Research Center, Yorktown Heights, New York
Education:	B.E.E. City College of New York, 1954 M.E.E. City College of New York, 1958 Ph.D., Polytechnic Institute of Brooklyn, 1966

Professional Activities:

President, IEEE Communications Society (1990-1991)

Vice President, IEEE Communications Society (1986-1988)

Chairman, Fellow Selection Committee, IEEE Communications Society (1983-1991)

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Chairman, National Commission C, URSI (1991-1992)

General Chairman, IEEE Communication Theory Workshop, Destin, 1996

General Chairman, Computer Communications Workshop, Reston, 1996

Co-General Chairman, IEEE Infocom '97, Kobe, Japan, 1997

Board of Governors, IEEE Communications Society (Elected Position) (1981-84) (1989-1993); Operational Committee; Chair. Nominations and Elections Committee

Fellow, IEEE

Fellow, American Association for the Advancement of Science

Fellow, Washington Academy of Sciences

Chairman, U.S. National Commission C, URSI (1990-92)

Elected to Executive Committee IEEE Washington Section (1973-74)

Elected to Executive Committee IEEE Northern Virginia Section (1983-1985)

General Chairman, Third Data Communication Symposium (IEEE/ACM)

Chairman, Technical Committee on Societal Implications of Technology (IEEE), 1984

Chairman, Professional Group on Information Theory Metropolitan Section (1971-72)

Member, Association for Computing Machinery (ACM)

Conference Board IEEE Communications Society (1980-83)

Member, Mathematical Association of America

Member, Commission C, URSI; U.S. delegate to XXth Congress

Delegate, National Research Council, National Academy of Science for IEEE (1982-84), Re-elected (1985-1987)

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Member, Washington Society of Engineers

Member, Philosophical Society of Washington

Member, Steering Committee of MILCOM, INFOCOM (1986-90)

Visiting Evaluator, Accrediting Board for Engineering and Technology (ABET) (1989-91)

External Evaluator, Graduate Programs, Northeastern University (1990)

Member, Scientific and Industrial Advisory Board. Premier's Technology Fund, Ottawa, Canada (1991 - 1996)

Scientific Advisory Board, Telecommunications Research Institute of Ontario (TRIO) (1990-1996) 1996).

Scientific Advisory Board, Institut National Researches de la Scientifique, Quebec, Canada, (1992-1996).

International Advisory Committee, TENCON, Melbourne, Australia, (1992).

General Chair, 22nd Communication Theory Workshop, Marathon, Florida.

Advisory Committee, International Symposium on Personal Communications, Nanjing, China, 1993.

Advisory Committee, Ministry of Industry, Trade and Technology, Ontario Centeres of Excellence, (1992-1993).

Review Panels (PYI, RIA and General) NSF, (1991-1997).

Reviewer (IEEE, ACM Journals, Wireless Networks, Electronics Letters, European Journal of Telecom, etc.).

International Advisory Board, TENCON, Singapore (1994).

General Chair, 25th Comunication Theory Workshop, Florida

Co-General Chair, INFOCOM'97, Kobe, Japan

General Chair, 11th Computer Communication Workshop, Reston, VA, Sept. 1996

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Editorships:

Editor, IEEE Transactions on Communications for Computer Communications (1972-76).

Associate Editor, Journal of Telecommunications Networks, (1981-1986).

Editor, Series of Communications and Signal Processing, Computer Science Press, Potomac, MD, (1981-).

Guest Editor, IEEE Transactions Special Issue on Computer Communications, Jan., 1977.

Guest Editor, IEEE Transactions Special Issue on Military Communications, Sept., 1980.

Guest Editor, IEEE Network, Special Issue on Network Security, April 1987.

Guest Editor, IEEE Communications Magazine, Special Issue on Technology, June 1987.

Guest Editor, IEEE Journal of Selected Areas of Communications, Special Issue on Secure Communications, 1989

Guest Editor, IEEE Journal on Selected Areas of Communications, Special Issue on Spread Spectrum Systems, 1990.

Guest Editor, IEEE Communications Magazine, Special Issue on Multimedia and on Social Issues, (?).

Editorial Board, Wireless Networks, J.C. Balzer, New Jersey, 1993 -

Honors and/or Honorary Fraternities:

Erskine Fellow (New Zealand) for 1997
Donald W. McLellan Award, 1994
IEEE Centennial Medal, 1984
Fellow, IEEE - "For contributions to the design of Digital Communications Systems and to Engineering Education", 1981
Fellow, American Association for the Advancement of Science (AAAS) for "Outstanding Research in Telecommunications", 1986

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Fellow, Washington Academy of Sciences, 1986
Elected to Cosmos Club, 1986
RCA Labs Research Award, 1954 - "For Outstanding Contributions to the Development of Color TV receivers"
Cosmos Club, 1986
ETA Kappa Nu
William Hance Memorial Medical in Mathematics
Rensselaer Polytechnic Institute Medal in Mathematics and Science
Pi Mu Epsilon Award for Interstate Mathematics Competition
Physics Teachers Award
Chemistry Teachers Award
Sigma Xi
Recipient of NSF Research Initiation Grant 1968-1969
Listings of Who's Who in America and Who's Who in American Men and Who's Who in Women of Science and Who's Who in Engineering (1989 – present)
Recipient of book award from the society of Technical Communications for "Local Area and Multiple Access Networks", 1986

Research Activities

Principal Investigator, NASA Grant on Space Communications, 1966-71.
Principal Investigator, NSF Grant on Modelling Analysis on Data Networks, 1971-76.
Co-Principal Organizer, NSF Symposium on Modelling and Analysis of Data Networks, 1975-76.
Principal Investigator, U.S. Army Research, Analysis of Development of Image Analysis for Computer Topography, 1979.
Principal Investigator, Joint IR & D, Melpar Division of E-Systems, Analysis and Design of Local Area Network, Array Processing Algorithms (1983-)
Principal Investigator, Joint IR&D, MCI Corp. (1989-93) Study of Fast Packet Switching.
Co-Principal Investigator, GLOMO (ARPA) 1996-97, Satellite Multimedia. Networks.
Co-Principal Investigator, Nortel 1997, CDMA Improvements.
Principal Investigator, INTELSAT (1989-1990) Study of Secure Command Lines for Satellite.
Currently doing research in packet switching, adaptive routing, and satellite communications, modelling of data networks, and Personal Communications Systems.
Also secure communications and electronic counter-counter measures; spread spectrum systems, adaptive null-steering antenna arrays, Microcellular Personal Communications Networks.

Principal Publications

"Statistical Decision Theory and Digital comm.," *ASEE Record*, June 1964, pp. 63-74.

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"Demodulation of Signals Transmitted Through a Random Channel," *PIBMRI Report* #1328-66, June 1966 (with M. Schwartz).

"A Recursive Approach to Signal Detection," *IEEE Trans. on Information Theory*, vol. IT-14, pp. 445-450, May 1968 (with R. Boorstyn).

"Transient Behavior of a Phase Locked Loop in the Presence of Noise" (with Dominiak) *IEEE Transactions on Communications Technology*, vol. Com. 18, No. 4, 1970 pp. 452-456.

"A Second-Order Gradient Algorithm", *Proc. of PIB Symposium on Computer Processing in Communications Technology*, August 1970.

"An Analysis of the Effectiveness of Hybrid Data Transmission Systems," *IBM Journal of Research and Development*, vol. 14, No. 4, July 1970, pp. 433-436.

"Terminal-Oriented Computer Communication Networks," *Proc. IEEE,* vol. 60, No. 11, November 1972, pp. 1408-1423 (with M. Schwartz and R. Boorstyn).

"Improvements in Routing in a Packet-Switched Network", *Proc. ICCC*, June 1974, pp. 249-252.

"Optimal Data Channel Equalization Using Normalized Stochastic Approximation Methods," *Proc. ISIT*, October 1974.

"Convergence Properties of Normalized Stochastic Approximation Methods," *Proc. IEEE on Decision on Control,* November 1974.

"Automatic Equalization Using a Successive Overrelaxation Iterative Technique", *IEEE Trans. on Information Theory,* January 1975.

"Currents in Computer Communication," in book *Computer Communication,* N. Macon, Ed., ICCC, Washington, D.C., 1975.

"Effects of a Priority Discipline or Routing in a Packet Switched Networks," (with McCoy), *IEEE Transactions on Communications,* vol. Com-24, March 1976, pp. 506-516.

"Analysis of a Reservation Multiple Access Technique for Data Transmission via Satellites," (with M. Balagangadhar), *IEEE Transactions on Communications,* vol. Com-27, No. 10, Oct. 1979, pp. 1467-1475.

Raymond L. Pickholtz
February 21, 1998

CURRICULUM VITAE

Dr. Branimir R. Vojcic,
Associate Professor of Engineering and Applied Science
Department of Electrical Engineering and Computer Science
The George Washington University
801 22nd Street, N.W., Washington, D.C. 20052

phone: (202)994-4874, fax: (202)994-0227
email: vojcic@seas.gwu.edu

Education

Doctor of Science Degree in Electrical Engineering (1989)
Thesis: "Performance of a class of digital transmission schemes in a fading dispersive channel with jamming", Faculty of Electrical Engineering, University of Belgrade, Yugoslavia.

Graduate Studies in Communications (1986/1987)
One year program, The George Washington University, Washington, DC.

Master of Science Degree in Electrical Engineering (1986)
Thesis: "Some possibilities of the ionospheric index of solar activity R_{12} prediction from the point of view of long term ionospheric radio wave propagation conditions prediction", Faculty of Electrical Engineering, University of Belgrade, Yugoslavia.

Dipl. Eng Degree in Electrical Engineering (1980)
Five year program, Faculty of Electrical Engineering, University of Belgrade, Yugoslavia.

Academic Experience

The George Washington University (from 1991 to present)
Associate Professor of Engineering and Applied Science in the Department of Electrical Engineering and Computer Science.

University of Belgrade (from 1988 to 1991)
Research Associate (part time)

Industrial Experience

Ministry of Defense, Yugoslavia (from 1981 to 1991, including leave of absence for studying)
Research engineer

Teaching at GWU:
- EE12 Signals and Systems (UG)
- EE143 Communications Engineering I (UG)
- EE144 Communications Engineering II (UG)
- EE148 Simulation of Communications Systems (UG/G)

- EE243 Communication Theory I (G)
- EE244 Communication Theory II (G)
- EE246 Digital Communications (G)
- EE247 Communications Systems (G)
- EE248 Computer Networks I (G)
- EE249 Computer Networks II (G)
- EE253 Mobile Communications (G)
- EE257 Spread Spectrum Communications (G)

Curriculum Improvements at GWU:
- Developed EE148, EE257
- Revised EE12, EE143, EE144

Course Directorship at GWU: EE12, EE148, EE246, EE247, EE253

Grants Received:
- *Wireless Information Network for Airport Surface Traffic Automation Using GPS* (11/1/92-12/31/92), PI, Martin Marietta.
- *Wireless Information Network for Airport Surface Traffic Automation Using GPS - Network Topology and Frequency Band Considerations* (2/1/93-12/31/93), PI, Martin Marietta.
- *Spectral Shaping for Improved Capacity in Multiple Access Communications* (12/1/93-11/30/94), PI, University Facilitating Fund.
- *Research in Simulation of Communications Systems* (5/1/94-12/31/94), PI, Elanix.
- *Research in Multiuser Detection and Innovative Approaches in Teaching* (7/1/95-6/30/98), PI, NSF Faculty Early CAREER Development Award.
- *Integration of DBS into Digital Battlefield using Commercial LEO Systems* (10/1/95-9/30/98), PI for GWU, joint proposal with CTA Incorporated (lead organization) and ALOHA Networks Inc. to ARPA (GLOMO Program).
- Research Experience for Undergraduates, Supplemental Grant from NSF (7/1/96-6/30/98),.
- *Multiuser Detection for IS-95 Air Interface* (1/1/97-12/31/97), NORTEL.

Software grants/gifts:
- SPW and BONeS (COMDISCO, Inc. now ALTA Group of CADENCE, Inc.)
- OPNET (MIL-3)
- SystemView (ELANIX, Inc.)

Graduated D.S. and M.S. Students:
- A. Ragab (D.S. in 1993): "Adaptive Equalization Performance in Mobile Fading Channels";
- S. Elagooz (D.S. in 1993): "Bandwidth Efficient Coding and Modulation for Time Dispersive Channels".
- D. Raushmayer (M.S. in 1995): "Analysis of the Capacity Region of Multiuser Channels".
- W.M. Jang (D.S. in 1996): "Joint Transmitter/Receiver Optimization in Synchronous Multiser Channels Over Multipath Channels";
- C. Hegarty (D.S. in 1996): "Noncoherent Multiuser Detection".
- Y. Shama (D.Sc. in 1997): "Detection/Decoding in Coded Multiuser Communications".

Professional Activities:

- Associate Editor of the IEEE Communications Letters
- Senior Member of IEEE.
- Past Vice Chairman, Communications Society of the Washington and Northern Virginia Chapter of IEEE.
- Chair, Technical Program, IEEE Communication Theory Workshop 1996, San Destin, FL.
- Member, Communication Theory Committee of the IEEE Communication Society.
- Member, Technical and Organizing Committee
  - ◊ International Symposium on Spread Spectrum Systems and Techniques 1988, Belgrade, Yugoslavia.
  - ◊ IEEE Communication Theory Workshop 1993, Marathon, FL.
  - ◊ Workshop on Mobility Management, GMU, Fairfax, VA, 1994.
- Member, Program Committee, Workshop on Mobility Management, Paris, 1996.
- Member, Program Committee, Workshop on Mobility Management, Melburne, 1997.
- Technical representative of the IEEE Communication Society to MILCOM'96, Reston, VA.
- Session Chair:
  - ◊ International Symposium on Spread Spectrum Systems and Techniques, 1988.
  - ◊ IEEE Communication Theory Workshop, 1993
  - ◊ MILCOM'96
  - ◊ Computer Communications Workshop, 1996
- Technical reviewer:
  - ◊ IEEE Transactions on Communications
  - ◊ IEEE Journal on Selected Areas in Communications
  - ◊ IEEE Transactions on Circuits and Systems
  - ◊ IEEE Transactions on Vehicular Technology
  - ◊ European Transactions on Telecommunications and Related Technologies
  - ◊ Wireless Personal Communications (An International Journal)
  - ◊ National Science Foundation
- Member of the Communications, Vehicular Technology and Signal Processing Societies and Information Theory Group of the IEEE.
- Member of the US Section of URSI.

Consulting:

- TV Answer
- NOVEL
- Motorola Satellite Communications
- CTA
- ORBCOM
- Lockheed Martin
- CBS

Awards:

- National Science Foundation Young Faculty Early CAREER Development Award in 1995.
- Radio TV Belgrade (Yugoslavia) Annual Award for the Best Paper in 1990 for the paper: B.R. Vojcic and R.L. Pickholtz, "Performance of coded direct sequence spread spectrum in a fading

dispersive channel with pulsed jamming", IEEE J. on Sel. Areas in Commun., vol. 8, NO. 5, pp. 934-942, 1990.

List of Publications

Book Contributions:

1. R.L. Pickholtz and **B.R. Vojcic**, "Issues in microcellular communications - CDMA versus TDMA", in *Worldwide Advances in Communications Networks*, edited by B. Jabbari, Plenum Press, New York, 1994.
2. R.L. Pickholtz and **B.R. Vojcic**, "CDMA for mobile LEO satellite communications", in *Code Division Multiple Access Communications*, edited by S.G. Glisic and P.A. Leppanen, Kluwer Academic Publishers, Norwell, MA, 1995.

Archival Journal Papers:

1. **B.R. Vojcic** and R.L. Pickholtz, "Performance of direct sequence spread spectrum in a fading dispersive channel with jamming", IEEE J. on Sel. Areas in Commun., vol. 7, No. 4, pp. 561-568, 1989.
2. **B.R. Vojcic** and R.L. Pickholtz, "Performance of coded direct sequence spread spectrum in a fading dispersive channel with pulsed jamming", IEEE J. on Sel. Areas in Commun., vol. 8, No. 5, pp. 934-942, 1990.
3. **B.R. Vojcic**, R.L. Pickholtz and L.B. Milstein, "Performance of DS-CDMA with imperfect power control operating over a low earth orbiting satellite link", IEEE Journal on Sel. Areas in Communications, vol. 12, pp. 560-567, May 1994.
4. **B.R. Vojcic**, R.L. Pickholtz and L.B. Milstein, "Total capacity in a shared CDMA LEOS environment", IEEE Journal in Sel. Areas in Communicationsin, vol.13, pp. 232-244, Feb. 1995.
5. **B.R. Vojcic**, R.L. Pickholtz and L.B. Milstein, "DS-CDMA outage performance over a mobile satellite channel", European Transactions on Telecommunications and Related Technologies, vol.6, No. 1, pp. 63-70, Jan.-Feb. 1995.
6. **B.R. Vojcic**, L.B. Milstein and R.L. Pickholtz, "Downlink DS CDMA performance over a mobile satellite channel", IEEE Trans. on Veh. Technology, vol. 45, No. 3, pp. 551-560, Aug. 1996.
7. V. Vanghi and **B. Vojcic**, "Multiuser detection with soft interference cancellation", Wireless Personal Communications (An International Journal), Special Issue on Signal Separation and Interference Cancellation for Personal, Indoor and Mobile Radio Communications , **3**: pp. 111-128, 1996.
8. W.M. Jang, **B.R. Vojcic** and R.L. Pickholtz, "Joint transmitter/receiver optimization in synchronous multiuser communications over multipath channels", IEEE Trans. on Commun, vol. 46, No. 2, pp. 269-278, Feb. 1998.
9. C. Hegarty and **B. Vojcic**, "Noncoherent two-stage multiuser detection of M-ary orthogonal signals", Wireless Networks Journal, August 1998.
10. **B.R. Vojcic** and W.M. Jang, "Transmitter precoding in synchronous multiuser communications", to appear in the IEEE Transactions on Commun. In 1998.
11. Y. Shama, **B.R. Vojcic** and B. Vucetic, "Suboptimum soft output detection algorithms for coded multiuser", to appear in the IEEE Trans. on Communications in 1998.

Appendix J

Appendix J

Audio Quality and Coding

There are several factors which affect the audio quality that can be achieved by a DAB system. Taking just one factor into account when deciding on an audio compression scheme and broadcast bit rate will not lead to an optimized IBOC DAB system.

First, an IBOC DAB system must allow a broadcaster to serve the same geographic area as that served by the existing analog broadcast. Second, within the coverage area, the digital signal must have high immunity to multipath, noise and interference. This "robustness" requires significant error concealment and protection against drop outs. Third, the digital signal must not noticeably increase noise in the analog host or in adjacent channels. Fourth, the codec should be based on commonly used coding schemes to promote standardization. Balancing all of these factors, USADR has selected MPEG Advanced Audio Coding ("MPEG AAC") as the audio compression scheme which optimizes its system performance with a 96 kbps rate for the FM system and 48, 32 or 16 kbps rates for the AM system.

MPEG AAC is the latest MPEG[1] standard on perceptual audio coding and is part of the world-wide MPEG family of audio and video standards.[2] Much of the work on AAC was done by Fraunhofer, AT&T, Dolby Labs, and Sony. These companies are leading experts in audio compression technology. It builds upon the existing MPEG Layer-3 standard by further optimizing coding efficiency.

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1 "MPEG" is the Moving Pictures Expert Group, working under the joint direction of the International Standards Organization ("ISO") and the International Electro-Technical Commission ("IEC"). Its main goal is the standardization of audio and video coding schemes.

2 MPEG AAC was standardized as ISO 13818-7 in April 1997. It is an outgrowth of the efforts of world leaders in audio technology. In addition to Fraunhofer, the patent pool for AAC includes technology developed at AT&T, Dolby Labs, Sony, and others.

"CD-quality" is defined as a 16 bits/sample Pulse Code Modulation ("PCM") linear system with a sampling rate of 44.1 kHz.[3] The sampling rate and number of bits supports an audio response of 20–22 kHz and a dynamic range of 96 dB. For a two channel stereo system this represents a data rate of 1,411,200 bits per second (2 channels * 16 bits * 44,100 = 1,411,200). AM and FM broadcasting channel bandwidths lack the capacity to support these data rates, thus audio compression technology is used to transmit the best sound possible in the available bandwidth.

Algorithms supporting audio compression are defined as perceptual audio coders and are referred to as codecs (coder - decoder). Like all perceptual audio coding schemes, MPEG AAC makes use of the signal masking properties of the human ear in order to reduce the amount of data that is transmitted. In any audio coder, the quantization noise is distributed to frequency bands in such a way that it is masked by the total signal (*i.e.*, it remains inaudible).

AAC is a very flexible coding scheme supporting sample rates from 8 kHz to 96 kHz. It can encode mono and stereo input data as well as multichannel data up to 48 channels. It is used for a wide range of applications, from Internet audio to multichannel surround sound. The high coding efficiency makes AAC attractive, especially for applications with very high quality demands or very limited transmission bandwidth.[4] Even though the basic structure of AAC is similar to previous audio coding techniques, including the commonly used MPEG Layer 3, AAC contains numerous innovations which are particularly helpful for the implementation of DAB.

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[3] When an audio compression scheme is integrated into a broadcasting system, block error rate is used as a metric in determining audio virtual CD-quality. A block error rate of 0.01 is conservatively considered "virtual" CD-quality.

[4] AAC has been chosen to be the audio coding standard for the Japanese HDTV system which will be introduced in 2000.

The crucial differences between MPEG AAC and its predecessor MPEG Layer-3 are shown as follows:

- *Filter Bank*: In contrast to the hybrid filter bank of MPEG Layer-3, which was chosen for reasons of compatibility but ultimately displayed certain structural weaknesses, MPEG AAC uses a plain Modified Discrete Cosine Transform ("MDCT"). Together with the increased window length (2048 instead of 1152 lines per transformation), the MDCT outperforms the filter banks of previous coding methods.
- *Temporal Noise Shaping ("TNS")*: A true novelty in the area of time/frequency coding schemes, TNS shapes the distribution of quantization noise in time by prediction in the frequency domain. Voice signals in particular experience considerable improvement through TNS.
- *Prediction*: This is a technique commonly used in the area of speech coding systems. It benefits from the fact that a certain type of audio signal is easy to predict.
- *Quantization*: By allowing finer control of quantization resolution, the given bit rate can be used more efficiently.
- *Bit-Stream Format*: The information to be transmitted undergoes entropy coding in order to keep redundancy as low as possible. The optimization of these coding methods together with a flexible bit-stream structure has made further improvement of the coding efficiency possible.

During the standardization process MPEG performed numerous listening tests to assess the audio quality of AAC. It is difficult to specify audio coded performance in terms of traditional audio measurement techniques such as frequency response, distortion, and dynamic range; therefore, audio codecs are psychoacoustically compared against a CD reference. In these double blind tests, human testers are given the opportunity to compare compressed and non-compressed segments of the same selection and make judgments as to the quality of the compressed segment. In tests designed to replicate the worst case signals, the AAC codec at 96 kpbs has shown to be almost indistinguishable from the original selection. For the most extreme cases, the difference in the compressed signal is audible, but not considered a major issue for listeners. This level of performance has not been achieved by any other CODEC at these low rates.[5] These tests use what is essentially a short audio clip played over and over to train the listener.[6] In other words, while listening to 96 kbps AAC encoded audio with high quality headsets, the average listener will not be able to distinguish between it and the original CD unless a short music selection is played over and over from both the CD and AAC (*i.e.*, the listener is trained). Because typical radio listeners will never listen to IBOC DAB under such pristine lab conditions with studio quality headsets and amplifiers, 96 kbps AAC will be perceived by typical listeners as "virtually" the same as a CD.

The top AM codec rate of 48 kbps will provide a significant improvement in audio over today's analog AM. Just as with the tradeoffs with FM, the AM channel and characteristics dictate this top rate. Where 48 kbps can be achieved, the audio will sound much like analog FM stereo without multipath and noise. Where 48 kbps cannot be achieved, the DAB system will

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[5] ISO Report on the MPEG-2 AAC stereo verification tests 2/98 Meares, Watanabe, Scheirer.

[6] *Id.*

operate at 32 kbps which will still support stereo, but high frequencies will be reduced. At 16 kbps, the audio will be equivalent to existing mono

There have been numerous improvements to codecs in the past decade. MPEG standards have made great strides in audio compression AAC is not backwards compatible with earlier MPEG standards, and can thus incorporate the latest research results without any compatibility restrictions. This allows AAC to be an effective codec choice for emerging applications, which makes it an excellent choice for DAB.

Appendix K

Blend With Time Diversity

Blend with time diversity is an effective method for ensuring graceful degradation of the digital audio, enhancing digital service to mobile receivers. It also provides a capability for rapid tuning and eliminates the need to compromise interleaver length. When a receiver is used in a mobile environment it is inevitable that the broadcast signal will be corrupted at some point. All listeners to analog radio have experienced a loss of signal due to natural or man-made obstructions or other impairments. Blend makes use of time diversity between two independent transmissions of the same audio source to reduce the probability of a corrupted signal.

As a digital receiver approaches the edge of coverage, its signal quality deteriorates abruptly. To compensate, the USADR IBOC DAB system uses blend as a means of gracefully degrading the signal as the edge of coverage is approached.

The USADR system uses blend with time diversity in both the hybrid and all-digital modes. The hybrid system uses the delayed analog signal as a backup; the all-digital system contains a delayed, low bit rate digital backup signal. In either case, when the primary digital signal is corrupted for a short time, the outage at the decoder is heard after the diversity delay. This diversity delay is incurred at the receiver and is comprised of deinterleaving and FEC decoding delay, audio decoding delay, and any additional delay for diversity improvement. The FEC decoder can be used to identify faulty audio frames; therefore, the exact time of the DAB audio outage can be predicted, signaling the receiver to blend from the primary to the backup signal. The listener may detect the temporary degradation in audio quality during the blend duration, but will not experience muting or undesirable artifacts.

Figure K-1 graphically depicts the blend with time delay process. The top part of the figure shows that, for the transmitted signal, the backup portion is delayed relative to the primary digital portion. This delay at the transmitter provides the time diversity as the signal travels through the channel. The top part of the figure also shows an impairment that prevents recovery of sections 6 and 7 of the digital signal. The impairment also interferes with sections 2 and 3 of the backup signal.

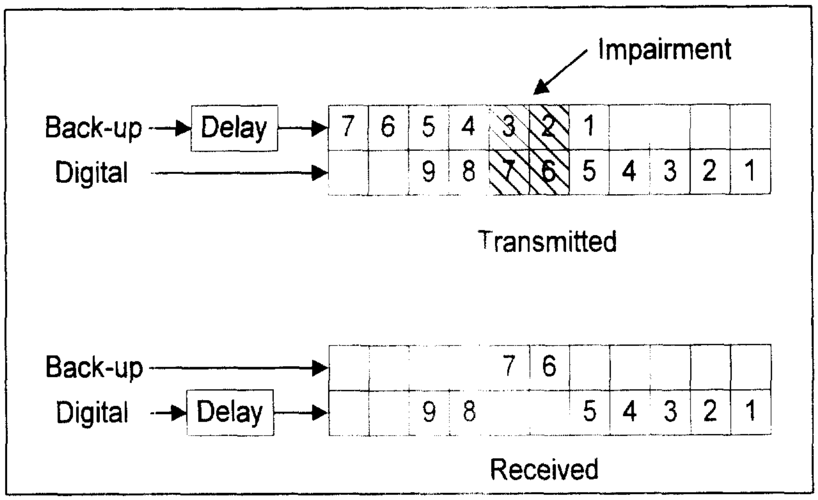

Figure K-1 - Blend Feature

The bottom part of Figure K-1 shows the received signal. At the receiver, the primary digital signal is delayed relative to the backup, and the signals are time-aligned. Part of the delay is due to the processing required to recover the primary digital signal, and the remainder of the delay can be implemented directly.

As shown in the bottom part of Figure K-1, Sections 1–5 of the digital signal are recovered. Sections 6 and 7 of the digital signal will be marked as non-recoverable by the receiver. However, because of the time-diversity, Sections 6 and 7 of the backup signal are not affected by the impairment, and the receiver can smoothly blend to the backup signal, with the blend beginning during Section 5. When the primary digital signal can be recovered again, the receiver can smoothly blend back to the primary digital signal.

If the diversity delay is sufficiently large that the primary signal and backup signal outages are independent, then the probability of an outage after diversity is the square of the probability of outage without diversity. For instance, if the probability of an outage is 1.0%, then the probability of outage after diversity is 0.01%, which is a great improvement. The actual performance can be quantified with knowledge of the autocorrelation function of the channel outage due to severe impairment. However, even without going into details of specific channel scenarios that may cause outages, it is clear that the blend function will considerably enhance the robustness of the system. The blend function also allows a means of gracefully degrading the digital audio as a receiver approaches the edge of coverage. When the receiver detects corrupted digital audio in its primary channel, it will blend in its delayed backup channel to provide uncorrupted audio.

The blend feature also solves the problem of fast tuning time. Without blend, a receiver would incur the diversity delay after tuning to a station before the listener hears the audio. The blend feature will demodulate the backup signal almost instantly, allowing the listener to hear the selection before blending to the primary digital signal several seconds later.

Appendix I

Auxiliary Services

DAB, in its most basic form, is simply a digital bitstream being transmitted in the AM and FM bands. Because that digital bitstream can be devoted to either audio or data, the IBOC DAB system has the capability to provide auxiliary services in addition to the primary audio signal. In order to achieve USADR's design requirement of improving robustness and audio quality, the majority of the capacity of the IBOC DAB system is devoted to the primary audio signal. At the same time, however, the system supports auxiliary services which will upgrade existing analog FM subsidiary communications services. The auxiliary services will be available in both the hybrid and all-digital modes of operation.

Current subcarriers or subsidiary communications authority ("SCA") broadcast with analog radio are used to provide a variety of services. SCAs may be used for audio or data services and require special receivers for use by the public. SCAs, although not widely used, are available for numerous niche services including paging, inventory distribution, bus dispatching, stock market reports, traffic control signal switching, point-to-point or multipoint messages, foreign language programming, radio reading services, radio broadcast data systems, station control and meter reading, utility load management, and muzak.[1] IBOC DAB auxiliary services will enhance the availability, reliability and robustness of the SCA-like services.

The USADR IBOC DAB system incorporates four types of data which can be used for auxiliary services:

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[1] *See Radio Subcarriers* (visited Sept. 28, 1998) <http://www.fcc.gov/mmb/asd/subcarriers/sub.html>.

1. Ancillary Services - Guaranteed real-time audio or data sent through a reserved channel, as opposed to a reserved subcarrier [2]

2. Opportunistic Data - This data-only service is multiplexed in with the audio during periods when the audio does not require full bandwidth, dynamic range or is not complex. In such cases, the audio does not require full throughput. Each message is assigned a priority and is transmitted based on that priority versus the complexity of the audio. Therefore, some latency is inevitable. This type of data intelligently recaptures capacity from the primary audio signal so as to minimize the impact to audio quality.

3. Supplementary Services - Audio or data placed in optional OFDM carriers that augment the primary audio signal OFDM carriers.

4. Secondary Auxiliary Services - Additional low power carriers are added in the all-digital mode, providing additional capacity for audio or auxiliary services.

FM DAB System:

USADR's FM system has a potentially large capacity that can be devoted to auxiliary services. The broadcaster can make adjustments to digital or analog audio quality and digital signal robustness depending on the desired auxiliary service capacity. An error-protected throughput of up to 104 kbps can be achieved in the hybrid mode. This breaks into the four categories listed above as follows:

1. Ancillary Services: Up to 32 kbps can be transmitted guaranteed real time by reducing the audio rate from 96 kbps to 64 kbps. Certain types of music and most

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[2] Both ancillary services and opportunistic data consume capacity from the same channel. Thus, greater levels of one will translate into less of the other.

voice broadcasts will not be affected by the audio rate reduction. The ancillary services rate can be set to 32 kbps with a corresponding reduction of audio rate.

2. Opportunistic Data: Up to 32 kbps can be sporadically multiplexed with the audio on a priority basis. Program Associated Data ("PAD"), which is used to display audio and station information in the receiver, takes a portion of the capacity since it is not required to be transmitted real time. As noted, the total of ancillary services and opportunistic data cannot exceed 32 kbps. Data is multiplexed at times when it will have minimal impact on audio quality.

3. Supplementary Services: Between 101 kHz and 129 kHz out from center channel, the FM system has the capacity to support 48 kbps of audio or data in the hybrid mode and 24 kbps in the all-digital mode.[3] Activating these supplementary carriers while in the hybrid FM mode may introduce additional noise into existing stereo analog receivers.

   An additional 24 kbps for Supplementary Services is available on the outer carriers from 185 to 199 kHz if the station can tolerate the interference from the first-adjacent analog stations onto these carriers. Although this latter option may be used in hybrid or all-digital modes, it is most attractive in the all-digital mode since the all-digital signal is virtually immune to all-digital first adjacent interferers.

4. Secondary Auxiliary Services: The all-digital mode includes 128 kbps of low power audio or data which exists between the two high-power digital sidebands. Although

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[3] The all-digital mode has lower capacity because 24 kbps of the capacity in this portion of the channel is required for the low bit rate backup digital signal needed for blend.